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EXAMINER	
SEFCHECK, GREGORY B	
ART UNIT	PAPER NUMBER
2662	

DATE MAILED: 02/23/2006

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	09/745,387	ORAN, DAVID R.
	Examiner	**Art Unit**
	Gregory B. Sefcheck	2662

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE _3_ MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on _29 November 2005_.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) _1-50_ is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) _1-50_ is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

- Applicant's Amendment filed 11/29/2005 is acknowledged.
- Claims 1, 10, 37, 47, and 48 have been amended.
- The previous objection to claim 47 has been withdrawn in light of the present amendments.
- The previous rejections of claims 1-12 under 35 USC 112, 2nd paragraph are withdrawn in light of the present amendments.
- Claim 50 has been added.
- Claims 1-50 remain pending.

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1, 2, 4, 6-9, 12, 24, 25, 27, 29-33, 35, 36, 41-45, and 47 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kung et al. (US006775267B1), hereafter Kung.

- In regards to Claims 1, 2, 24, 25, 33, 36, and 45,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system (Figs. 1-4; Col. 7, lines 15-25; Col. 8, lines 9-13; claim 1,24,36 – method, system, and electronic storage medium having software for controlling a VOIP call).

Kung shows that VOIP call packets traveling through the IP network may be given a priority to maintain certain QoS requirements (Col. 7, lines 21-25; claim 1,24,36 – tracking adaptation schemes used for transmitting packet in a VOIP call).

Kung discloses the ability to change quality of service, required bit rate, priority, etc. in real time in response to user input (Col. 7, lines 27-30; claim 1,24,36 – monitoring a user response/input that requests a different level of user perceived sound quality for the VOIP call; claim 1,24,36 – dynamically varying the adaptation schemes used for transmitting the packets in the call to correspond with the requested level of quality).

Kung further discloses that the real time changes to the VOIP call may be flexibly performed with regard to congestion in the network (Col. 7, lines 30-35; Col. 17, lines 55-59; claim 33,45 – monitoring congestion in a network used for conducting the call and varying adaptation schemes according to the user response and the monitored congestion).

Kung discloses that calls may be initially conducted at a user's default settings of quality, cost, etc. (Abstract; Col. 28, lines 12-19; claim 2,25 – initially transmitting packets of VOIP call using best effort).

The call settings may then be dynamically altered based on user input, requiring a call manager to reserve the necessary resources (Col. 30, lines 25-30; claim 1 – dynamically varying adaptation schemes effects how much digital data is used to

represent an audio signal; claim 2,25 – monitoring the user response for a request to increase sound quality; claim 2,25 – requesting reservation of resources during the call when the increased sound quality request is detected prior to the reserved resources being used during the call and without necessarily using the entire requested resources during the call).

Kung does not explicitly show controlling the VOIP call, including tracking and dynamically varying the adaptation schemes at a telephone endpoint.

However, Kung discloses that the system is flexible so that a given communication can be dynamically altered according to *customer preferences* (emphasis added) such as a user's desired quality of service. As such, the adaptation schemes are tracked and varied by the user (telephone endpoint), though the user's input is processed at the central station (Col. 7, lines 27-35; claim 1,24,36 – tracking and varying of adaptation schemes at a telephone endpoint). Furthermore, the central station utilized in Kung enables the above functionality to be provided to a plurality of user endpoints through a single entity. While inefficient in the multi-user domain of Kung, one of ordinary skill in the art at the time of the invention could certainly perform the functionality of the central station per user endpoint if such functionality were to only be provided to particular users.

It would have been obvious to one of ordinary skill in the art at the time of the invention to incorporate the functionality of the central station in Kung to a track and vary adaptation schemes at a particular telephone endpoint of the system. This would

enable only select telephone endpoints to be able to track and vary adaptation schemes based on the capabilities of that particular endpoint.

- In regards to Claims 4 and 27,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Kung shows that user input for changing the call quality may be performed before as well as during the call (Col. 7, lines 27-35; Col. 30, lines 25-30; claim 4,27 – conducting the already established call using reserved resources when the reservation request is accepted and the user response requests additional increases in sound quality).

- In regards to Claims 6, 8, 29, 31, 41, and 43,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Kung discloses changing call parameters is accomplished through user input on a user device, such as the screen portions shown in Figs. 7-9. Kung discloses that user input may be collected via touchscreen (graphical user interface; Col. 20, lines 51-55; claim 6,29,41 – using a signal generated by an input device to detect the user response during the call; claim 8,31,43 – using a graphical user interface as the input device).

- In regards to Claims 7, 9, 30, 32, 42, and 44,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Telephone units are also shown to be connected to the system for use as an input device by the user, including DTMF sensing logic (Fig. 3; Col. 23, lines 45-51; claim 7,30,42 – including using a dial or buttons on a telephone as the input device; claim 9,32,44 – including decoding DTMF signals to detect the user response).

- In regards to Claims 12, 35, and 47,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Referring to Fig. 9B, Kung shows that user input for changing call parameters may include cost icons (claim 12,35,47 – detecting a user response selecting a cost for the VoIP call and varying the adaptation schemes according to the selected cost).

3. Claims 3 and 26 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kung in view of Karagiannis (US20020015395A1).

- In regards to Claims 16 and 38,

Kung discloses a system for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Kung does not explicitly disclose utilizing an RSVP request during the call to request reservation of resources.

Karagiannis discloses route optimization (Title) in which RSVP protocol is used for requesting and reserving network resources for a VOIP call (Figs. 3-6; Abstract; claim 16,38 – requesting reservation of resources comprises making RSVP request during the call).

It would have been obvious to one of ordinary skill in the art at the time of the invention to utilize RSVP requests for reserving resources, as shown by Karagiannis, during a VOIP call in the system of Kung. This modification would enable a bandwidth reservation request for the call to specify certain quality of service requirements needed to improve the sound quality solicited by a user in Kung.

4. Claims 5, 10, 11, 13-15, 17-23, 28, 37, 39, 48, and 50 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kung in view of Havens (US006735175B1).

- In regards to Claims 5, 11, 28, and 50,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Kung does not explicitly disclose increasing voice coder performance or reducing payload size after the resources are reserved.

Havens discloses changing quality of service for voice over IP calls. Havens shows implementing requested changes to quality of service by adjusting performance

of the codec module (Fig. 2; Abstract; Col. 2, lines 31-43; Col. 4, lines 23-30; claim 5,28 – increasing voice coder performance or reducing payload size after the resources are reserved; claim 11 – varying codecs used for encoding audio signals into digital data making up the packets). Havens shows that the call-originating user dials a change of QoS on the telephone when the perceived audio quality of the call is insufficient. Havens shows that these changes may be done in real time during the call (Col. 4, lines 12-53; claim 50 – listening to audible signal after dynamically varying adaptation schemes to determine level of user perceived sound quality, further varying the schemes to improve the audible signal when the perceived sound quality is low, and further listening to the improved signal to determine a level of user perceived sound quality).

It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method and system of Kung by adjusting coder performance in response to user requested change in quality of service, as shown by Havens. This would enable quality of service to be dynamically adjusted during a call without requiring changes to the bandwidth of the call.

- In regards to Claims 10, 13-15, and 37,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system (Figs. 1-4; Col. 7, lines 15-25; Col. 8, lines 9-13; claim 10,37 – method, system, and electronic storage medium having software for controlling a VOIP call).

Kung shows that VOIP call packets traveling through the IP network may be given a priority to maintain certain QoS requirements (Col. 7, lines 21-25; claim 10 – tracking adaptation schemes used for transmitting packet in a VOIP call).

Kung discloses the ability to change quality of service, required bit rate, priority, etc. in real time in response to user input (Col. 7, lines 27-30; claim 10,13,37 – monitoring a user response/input that requests a different level of user perceived sound quality for the VOIP call; claim 10,13,37 – dynamically varying the adaptation schemes used for transmitting the packets in the call to correspond with the requested level of quality).

Kung further discloses that the real time changes to the VOIP call may be flexibly performed with regard to congestion in the network (Col. 7, lines 30-35; Col. 17, lines 55-59; claim 10,14 – monitoring congestion in a network used for conducting the call and varying adaptation schemes according to the user response and the monitored congestion).

Kung discloses that calls may be initially conducted at a user's default settings of quality, cost, etc. (Abstract; Col. 28, lines 12-19; claim 15,37 – initially transmitting packets of VOIP call using best effort).

The call settings may then be dynamically altered based on user input, requiring a call manager to reserve the necessary resources (Col. 30, lines 25-30; claim 13,37 – dynamically varying adaptation schemes changes how an analog signal is converted into packets of the call; claim 37 – monitoring the user response for a request to increase sound quality; claim 15,37 – requesting reservation of resources during the call

when the increased sound quality request is detected prior to the reserved resources being used during the call and without necessarily using the entire requested resources during the call).

Though Kung shows that quality of service control through a multimedia gateway control protocol may include codec choice (Col. 14, lines 59-65), Kung does not explicitly disclose dynamically varying adaptation schemes as including either varying which coder algorithm is used, varying a packet payload size, or varying the type of FEC used.

Havens discloses changing quality of service for voice over IP calls. Havens shows implementing requested changes to quality of service by adjusting performance of the codec module (Fig. 2; Abstract; Col. 2, lines 31-43; Col. 4, lines 23-30; claim 10 – varying which coder algorithm is used).

It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method and system of Kung by adjusting coder performance in response to user requested change in quality of service, as shown by Havens. This would enable quality of service to be dynamically adjusted during a call without requiring changes to the bandwidth of the call.

Kung does not explicitly show varying the coder algorithm used at a telephone endpoint.

However, Kung discloses that the system is flexible so that a given communication can be dynamically altered according to *customer preferences* (emphasis added) such as a user's desired quality of service. As such, the adaptation schemes are tracked and varied by the user (telephone endpoint), though the user's input is processed at the central station (Col. 7, lines 27-35; claim 1,24,36 – tracking and varying of adaptation schemes at a telephone endpoint). Furthermore, the central station utilized in Kung enables the above functionality to be provided to a plurality of user endpoints through a single entity. While inefficient in the multi-user domain of Kung, one of ordinary skill in the art at the time of the invention could certainly perform the functionality of the central station per user endpoint if such functionality were to only be provided to particular users.

It would have been obvious to one of ordinary skill in the art at the time of the invention to incorporate the functionality of the central station in Kung to a track and vary adaptation schemes at a particular telephone endpoint of the system. This would enable only select telephone endpoints to be able to track and vary adaptation schemes based on the capabilities of that particular endpoint.

- In regards to Claims 17 and 39,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Kung shows that user input for changing the call quality may be performed before as well as during the call (Col. 7, lines 27-35; Col. 30, lines 25-30; claim 17,39 –

conducting the already established call using reserved resources when the reservation request is accepted and the user response requests additional increases in sound quality).

- In regards to Claims 18 and 21,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Telephone units are also shown to be connected to the system for use as an input device by the user, including DTMF sensing logic (Fig. 3; Col. 23, lines 45-51; claim 18 – including using a dial or buttons on a telephone as the input device; claim 21 – including decoding DTMF signals to detect the user response).

- In regards to Claims 19,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Kung discloses changing call parameters is accomplished through user input on a user device, such as the screen portions shown in Figs. 7-9. Kung discloses that user input may be collected via touchscreen (graphical user interface; Col. 20, lines 51-55; claim 19 – using a graphical user interface as the input device).

- In regards to Claim 20,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Referring to Fig. 9B, Kung shows that user input for changing call parameters may include cost icons (claim 20 – detecting a user response selecting a cost for the VoIP call and varying the adaptation schemes according to the selected cost).

- In regards to Claims 22 and 23,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Referring to Figs. 7-9, Kung shows that user input determines how much the call parameters are varied (claim 22 – user response determines how much the adaptation parameters are varied).

Kung discloses that data rate is one of the call parameters that may be varied in response to user input (Abstract; Figs. 7b and 9c; Col. 7, lines 25-30; claim 23 – varying the rate packets are transmitted and received during the call).

- In regards to Claim 48,

Referring to Fig. 1, Kung shows that a call from PSTN 160 may interface IP network 120 through a gateway, where it would be converted to a packetized call (claim 48 – establishing a call over POTS/PSTN; claim 48 – packetizing the call at a network device connected to a packet network).

Kung discloses the ability to change quality of service, required bit rate, priority, etc. in real time in response to user input (Col. 7, lines 27-30). Telephone units are shown to be connected to the system for use as an input device by the user, including DTMF sensing logic (Fig. 3; Col. 23, lines 45-51; claim 48 – generating DTMF signal to request modification of sound quality; claim 48 – detecting the DTMF signals and modifying the adaptation parameters to modify the sound quality).

5. Claims 16, 38, and 40 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kung in view of Havens as applied to claims 15 and 37 above, and further in view of Karagiannis (US20020015395A1).

- In regards to Claims 16 and 38,

Kung discloses a system for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Kung does not explicitly disclose utilizing an RSVP request during the call to request reservation of resources.

Karagiannis discloses route optimization (Title) in which RSVP protocol is used for requesting and reserving network resources for a VOIP call (Figs. 3-6; Abstract; claim 16,38 – requesting reservation of resources comprises making RSVP request during the call).

It would have been obvious to one of ordinary skill in the art at the time of the invention to utilize RSVP requests for reserving resources, as shown by Karagiannis,

during a VOIP call in the system of Kung. This modification would enable a bandwidth reservation request for the call to specify certain quality of service requirements needed to improve the sound quality solicited by a user in Kung.

- In regards to Claim 40,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Kung does not explicitly disclose increasing voice coder performance or reducing payload size after the resources are reserved.

Havens discloses changing quality of service for voice over IP calls. Havens shows that implements requested changes to quality of service by adjusting performance of the codec module (Fig. 2; Abstract; Col. 2, lines 31-43; Col. 4, lines 23-30;).

It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method and system of Kung by adjusting coder performance in response to user requested change in quality of service, as shown by Havens. This would enable quality of service to be dynamically adjusted during a call without requiring changes to the bandwidth allocated to the call and elsewhere within the system.

6. Claims 34 and 46 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kung in view of Havens as applied to claim 11 above, and further in view of Rosenberg et al. (US006141788A), hereafter Rosenberg.

- In regards to Claims 34 and 46,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims. Kung further discloses that a change in data rate may be performed in response to user input (Abstract; Figs. 7b and 9c; Col. 7, lines 25-30; claim 34,46 – varying the rate packets are transmitted and received during the call).

Kung does not explicitly disclose varying codecs used for encoding audio signals into digital data making up the packets, varying an amount of audio data in the audio packets and adding or removing error correction information from the audio packets.

Havens discloses changing quality of service for voice over IP calls. Havens shows that implements requested changes to quality of service by adjusting performance of the codec module and the amount of data sampled for packet production (Fig. 2; Abstract; Col. 2, lines 1-11 and 31-43; Col. 4, lines 23-30; claim 34,46 – varying codecs used for encoding audio signals into digital data making up the packets; claim 34,46 – varying an amount of audio data in the audio packets).

Rosenberg discloses forward error correction in packet networks. Rosenberg discloses a method in which the degree of error correction included in a packet can be dynamically adjusted by the sender (Abstract; claim 34,46 – adding or removing error correction information from the audio packets).

It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method and system of Kung by varying codecs used for

encoding audio signals into digital data making up the packets and/or varying an amount of audio data in the audio packets, as shown by Havens, and dynamically adjusting (adding/removing) the amount of error correction information from the audio packets, as shown by Rosenberg. Each of these actions, performed alone or in combination, would enable an improvement in quality of service for a VOIP call without requiring changes to the bandwidth allocated to the call and elsewhere within the system.

7. Claim 49 is rejected under 35 U.S.C. 103(a) as being unpatentable over Kung in view of Kato (US005844918A).

- In regards to Claim 49,

Kung discloses a method for controlling a voice-over-IP (VOIP) call (Figs. 1-4; Col. 7, lines 15-25; Col. 8, lines 9-13; claim 49 – method for controlling a VOIP call).

Kung shows that VOIP call packets traveling through the IP network may be given a priority to maintain certain QoS requirements (Col. 7, lines 21-25; claim 49 – tracking adaptation schemes used for transmitting packet in a VOIP call).

Kung discloses the ability to change quality of service, required bit rate, priority, etc. in real time in response to user input (Col. 7, lines 27-30; claim 49 – monitoring a user response that requests a different level of user perceived sound quality for the VOIP call; claim 1,24,36 – dynamically varying the adaptation schemes used for transmitting the packets in the call to correspond with the requested level of quality).

Kung does not explicitly disclose adjusting FEC and packet payload length as included in the dynamically varying adaptation parameters.

Kato discloses a digital transmission/receiving method and apparatus. Kato discloses that adjustments to FEC and packet length impact the quality of transmission/reception in a system.

It would have been obvious to one of ordinary skill in the art at the time of the invention to enable adjustments to the FEC and packet length of a transmission as part of the dynamically varying adaptation schemes of Kung, as shown by Kato, because the FEC and packet length of a transmission effect the quality of a transmission. Therefore, adjustments to the FEC and packet length could result in quality changes requested by a user in Kung.

Response to Arguments

8. Applicant's arguments with respect to claims 10, 13-15, 17-23, 37, 39, and 48 have been considered but are moot in view of the new ground(s) of rejection.

9. Applicant's arguments with respect to claims 3, 16, 26, and 38 have been considered but are moot in view of the new ground(s) of rejection.

10. Applicant's arguments filed 11/29/2005 have been fully considered but they are not persuasive.

- In the Remarks on pg. 11-12 and 14 of the Amendment, Applicant contends that there is no motivation to modify the prior art to meet the missing limitations in Kung. Furthermore, Applicant has mischaracterized the Examiner's comments from the November 16, 2005 telephone interview, in which Applicant contends the Examiner acknowledged a lack of motivation to modify Kung due to differing definitions of Quality of Service between the present application and the cited prior art of Kung, Havens and Kato.
- The Examiner respectfully disagrees. In the 11/16/2005 interview, the Examiner only acknowledged that the term "Quality of Service", or QoS, can have a wide range of meanings. Kung's disclosure refers to QoS of network resources, focusing mainly on cost-based path selection through the network and reliability of transmitted packets. However, thorough examination of Kung shows that signaling from a user through the network can specify default "QoS" for a broad range of settings, including codec choice (Col. 14, lines 60-65). While not explicitly disclosing the varying of coder algorithm as claimed, the suggestion and motivation to combine Kung with the teachings of Havens and Kato, as shown in the rejection above, is properly provided in the prior art.

- In the Remarks on pg. 12-13 of the Amendment, Applicant contends the motivation to modify Kung by tracking and varying adaptation schemes at

telephone endpoints rather than within the network is improper. Applicant further contends that Kung teaches away from such a modification.

- The Examiner respectfully disagrees. Modification of Kung for tracking and varying schemes at telephone endpoints, while inefficient for deployment to every network user, would be proper on a per user basis dependent on the particular schemes being tracked and varied. QoS settings such as codec choice impact only the specified user, as opposed to cost-based path selection, which impacts the entire network. Therefore, tracking and varying adaptation schemes with respect to coding algorithm would be more efficiently implemented at the telephone endpoints of Kung rather than within the network. Furthermore, Applicant's contention that Kung teaches away from the proposed modification is inaccurate. Disclosed examples and preferred embodiments do not constitute a teaching away from a broader disclosure or nonpreferred embodiments. The specific disclosure of Kung relating to the tracking and varying of adaptation schemes within the network does not constitute a teaching away from tracking and varying at the telephone endpoints because Kung does not criticize, discredit, or otherwise discourage the claimed solution. See MPEP 2123.

Conclusion

11. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Gregory B. Sefcheck whose telephone number is 571-272-3098. The examiner can normally be reached on Monday-Friday, 8:00am-4:30pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Seema Rao can be reached on 571-272-3174. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

GBS
2-17-2006

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